ATK Reaper: The Ambisonic Toolkit as JSFX plugins

Trond Lossius
BEK - Bergen Centre for Electronic Arts
`trond.lossius@bek.no`

Joseph Anderson
DXARTS, University of Washington
`joanders@uw.edu`

ABSTRACT

While there is a well-established workflow for stereo production in DAWs, options have been more limited when working with Ambisonics. The Ambisonic Toolkit (ATK) brings together a number of tools and transforms for working with first order Ambisonic surround sound, and includes intriguing possibilities for spatial soundfield imaging. These tools have previously only been available for public release via the SuperCollider real-time processing environment.

Cockos Reaper is a reasonably priced and flexible DAW, popular among many composers and sonic artists working with spatial sound. Reaper's versatile design conveniently supports the ATK's Ambisonic workflow model. Using the JSFX text-based scripting language, the ATK has now been ported to plugins for Reaper; these include intuitive graphical user interfaces.

1. INTRODUCTION

1.1 Spatial sound representations

Spatial sound is usually represented in one of three ways: as channel feeds, as spatial scene descriptions, or as soundfield encodings. ITU 5.1 and ITU 7.1 are examples of fixed-channel distribution formats [1]. This approach becomes less practical as the number of channels increases, and also has clear limitations due to the lack of flexibility in loudspeaker positioning. Object-based spatial scene descriptions instead associate sound sources with meta-information describing location or direction. In this way an auditory scene can be described independently of loudspeaker setup, and rendered appropriately for a chosen playback system. This is the approach used for e.g., Dolby Atmos and Wave Field Synthesis [2, 3]. SpatDIF, the Spatial Sound Description Interchange Format, is an open format offering a semantic and syntactic specification for storing and transmitting such spatial audio scene descriptions [4].

Ambisonics encodes audio sources into a speaker-independent representation of the soundfield called B-format [5]. Decoding is the process where an encoded soundfield is translated to individual speaker channel feeds; an advantage being, decoders can be designed for different speaker arrays. Ambisonics is based on spherical harmonic decom-

position of the soundfield, and depending on truncation of the spherical harmonic decomposition, the B-format signal may be first order (FOA) or higher order Ambisonic (HOA) [1] [6]. Spatial resolution improves with increasing order, but so does the number of channels required for the encoded signal. A number of commercial microphones are available for recording in FOA [7], and free or commercial B-format recordings are available as sound effects libraries. [2]

1.2 Spatial sound processing in DAWs

Encoding, processing and decoding of B-format signals requires support for multi-channel signals. Real-time programming environments are well-suited in this respect as they offer flexible configuration and routing of channels. Ambisonics is supported in all major real-time audio programming environments, such as Csound, Max, Pd and SuperCollider, often as third party extensions to these programs.

The support for surround sound is more limited in most Digital Audio Workstation (DAW) programs. Most DAWs that support surround sound, such as Adobe Audition, Cubase, Logic Pro X and Pro Tools are oriented towards fixed-channel distribution formats, with an upper limit of 6 (5.1) or 8 (7.1) channels. Digital Performer also supports 10.2, and Nuendo supports up to 13 channels in various fixed-channel distribution formats. Nuendo and Pro Tools can be extended to support object-based spatial scene descriptions. The Iosono Spatial Audio Workstation program plug-in for Nuendo extends the capabilities of this program by adding abilities for object-based manipulation of sound sources [8], and can be used with Iosono Core hardware systems for wave-field synthesis. Dolby Atmos authoring is achieved using ProTools and the Dolby Rendering and Mastering Unit (RMU). RMU provides the rendering engine for the mix stage, and integrates with Pro Tools through the Dolby Atmos Panner plug-in over Ethernet for metadata communication and monitoring. The metadata is stored in the Pro Tools session as plug-in automation [9].

The DAWs discussed so far are either limited with respect to surround sound abilities, or expensive and proprietary software and hardware systems for object-based spatial scene authoring and rendering. In contrast Reaper is a reasonably priced and flexible DAW for spatial sound. [3]

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[1] Third order Ambisonic is often abbreviated TOA.

[2] http://ambisonia.com, http://www.surround-library.com and http://www.spheric-collection.com. (All URLs in this article were last accessed July 10th 2014.)

[3] http://www.reaper.fm

Supporting tracks of up to 64 channels, [4] it is exceptionally open-ended with respect to routing of both channels and tracks. All standard fixed-channel surround sound configurations are supported, and the included *ReaSurround* panning plugin caters for non-standard speaker configurations [10]. Flexible routing makes Reaper well-suited for Ambisonics; it is a preferred DAW for many composers within the Ambisonic community. [5] Additionally, Reaper's *FX Chains* feature conveniently allows plugin chains to be saved as presets.

1.3 Plugins for Ambisonic processing

In recent years a number of plugins have emerged for Ambisonic processing. *WigWare* is a set of VST plugins for Mac and Windows that includes FOA and HOA panners and decoders for regular and irregular speaker layouts, as well as an FOA 3D reverb. [6] Courville has developed a suite of FOA, 2nd order and TOA plugins [7] . These include encoding and decoding, rotations, reverb, and additional convenient utilities. Additionally encoders and decoders for 5th order pantophonic [8] are also available. Unfortunately, the plugins are developed using SonicBirth [9] , and due to the long halt in development of this framework, these plugins do not work reliably in all newer DAW versions. *ambiX* are cross-platform FOA, TOA and 5th order Ambisonic processors for encoding and decoding, including binaural decoding, as well as spatial transforms [11, 12]. These plugins use the ambiX encoded signal convention, ACN channel ordering with SN3D normalization [13], but also provides a conversions utility for other formats. They are accompanied by *mcfx*, a number of more general multichannel plugins for equalising, delay, gain adjustment and level metering [14]. *Flux Ircam Spat* wraps the Ircam Spatialisateur multiformat room acoustics simulation and localization software as AU and VST plugins for Mac and Windows [15]. Although this plugin mainly provides object-based scene descriptions, it can render the auralization as B-format. The *HARPEX* plugin for decoding FOA uses a spatial upsampling algorithm based on high angular resolution planewave expansion (HARPEX) in order to expand the smaller sweet spot of FOA to HOA extents [16]. Blue Ripple Sound offers a wide range of OSX and Windows VST plugins for TOA, [10] including encoders, transcoders and decoders, and a rich set of plugins for spatial filtering of the encoded signal. Moreover, TOA includes a HARPEX upsampler, facilitating FOA to TOA conversion via the HARPEX algorithm.

2. AMBISONIC TOOLKIT

The Ambisonic Toolkit (ATK) brings together a number of tools and transforms for working with Ambisonic surround sound [17]. Use is targeted towards the composer of acousmatic and experimental music. The intention is for the toolset to be both ergonomic and comprehensive, providing algorithms to creatively manipulate and synthesize Ambisonic soundfields.

By focusing on the problem of synthesising and processing soundfields, the tools are framed for the user to 'think Ambisonically'. The model of the ATK is a *sound-field sound-image* model rather than a sound-object sound-scene model. In addressing the holistic problem of creatively controlling a complete soundfield, the ATK allows and encourages the composer to think beyond the placement of sounds in a sound-space and instead attend to the impression and image of a soundfield. This is viewed to be the *idiomatic* approach for working with Ambisonic technique, leveraging the model the technology presents.

The ATK has existed in a variety of forms since 1998, beginning as a collection of Csound orchestras, and later as a set of privately distributed VST plugins. Development of the real-time ATK library for SuperCollider2 began in 2000, and in recent years ATK has primarily been distributed as a version for SuperCollider3 [17]. Some of the underlying ideas of ATK has also been incorporated into the Blue Ripple Sound TOA plugins.

2.1 Ambisonic Toolkit Paradigm

Figure 1 illustrates how the ATK separates the task of production work with Ambisonics into three distinct elements:

Author: Capture or synthesise an Ambisonic soundfield.

Image: Spatially filter an Ambisonic soundfield.

Monitor: Playback or render an Ambisonic soundfield.

In its most simple form, Ambisonics can be regarded as splitting the panning law into two separate parts: *encoding* (Authoring) and *decoding* (Monitoring), where final panning (*decoding*) is deferred to an actual loudspeaker array at the time of audition. The ATK considers Imaging (*transforming*) a soundfield to be an critical step; this is where the artist shapes and processes the soundfield in a coherent way which isn't easily available via the other models for working with spatial sound. Further details are discussed in section 4, below.

Many publicly distributed implementations of Ambisonics provide only *encoding* and *decoding*. While giving flexibility regarding final playback, failing to include *transformers* misses out the concept of *imaging* and fails to capitalise on the advantages of the *sound-field sound-image* paradigm intrinsic to Ambisonics.

3. ATK REAPER: DESIGN CONSIDERATIONS

3.1 Choice of plugin architecture

When porting ATK for use in a DAW environment the prime question is whether to implement plugins in C++ using plugin architectures such as VST [11] or AudioUnit

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[4] Suitable for 7th order HOA.
[5] http://www.brucewiggins.co.uk/?page_id=215
[6] http://www.brucewiggins.co.uk/?page_id=78
[7] http://www.radio.uqam.ca/ambisonic/b2x.html
[8] Horizontal only.
[9] http://sonicbirth.com/
[10] http://www.blueripplesound.com/product-listings/pro-audio

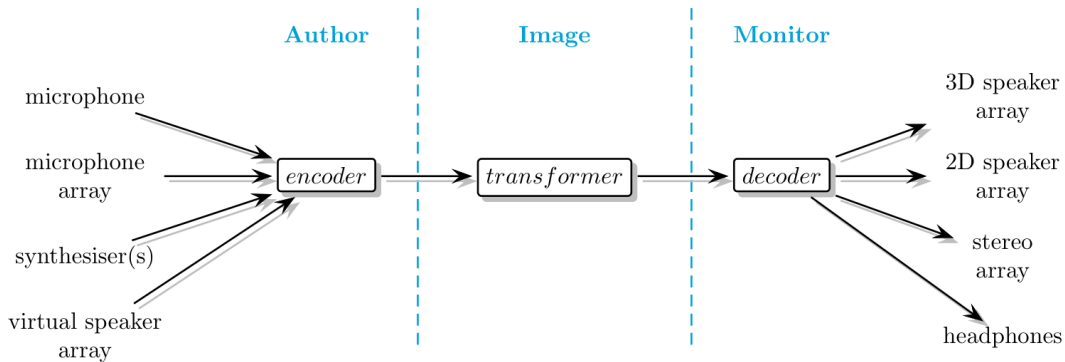[11] http://www.steinberg.net/en/company/developer.html

Figure 1: Ambisonic Toolkit Paradigm.

[18]. If doing this, it seems beneficial to use a intermediate library such as Faust [19], Juce [12] or wdl-ol [13], but it will still require substantial overhead in terms of having to compile and test the plugins for multiple processors and platforms. At the same time the plugins are vulnerable to future changes in OS and plugin platforms.

JSFX is a text-based scripting language for programming audio-oriented effects compiled on the fly for Cockos Reaper. [14] Using JSFX for Reaper there is no need to compile, and plugins immediately work with all platforms and processors supported by Reaper. Maintenance of the plugins are not expected to be demanding, as Reaper itself will take care of future changes to underlaying architectures. Using JSFX has been speeding up the development process substantially, and development has been able to primarily focus only on the specific processing and interfacing. One disadvantage is that graphics in plugins with custom GUIs seem to be demanding on system resources, and the program indicate major CPU overhead when plugin GUIs are open. However, once the GUI window is closed, the plugins do not seem to require any extra processing resources.

3.2 Coordinate systems and encoding conventions

Within the spatial audio community several coordinate system conventions are being used in parallel. Theory on Ambisonics generally assume the same coordinate system conventions as acoustics, with the x-axis pointing forward, y-axis to the left, and the z-axis pointing upwards. Spherical coordinates also follows standard mathematical conventions, with $0°$ azimuth being forward, and angles increasing in the anti-clockwise direction so that $90°$ azimuth is to the left. Positive elevation is upwards. This is the coordinate system convention used by ATK for SuperCollider

SpatDIF use a navigation-based coordinate system with x-axis to the right and, y-axis pointing forward, and z-axis pointing upwards. Azimuth is $0°$ in the forward direction, and angles increase clockwise so that $90°$ azimuth is to the right. Positive elevation is upwards [4].

There are also several conventions for how encoded Ambisonic signals are represented. B-format recordings made with one type of encoding and played back using another

will have severe mismatches in amplitude levels and channel order.

Classic *Furse-Malham* encoding (FuMa) can be used for up to third order signals, and at present appears to be the preferred representation in musical applications. FuMa format signals adhere to acoustics coordinate conventions. The W component is the pressure (omni or mono) component of the signal, X is the pressure gradient component in the forward-backwards direction, Y is the pressure gradient component in the left-right direction and Z is the up-down pressure gradient component. FuMa encoding is the preferred format for all of the Ambisonic plugins discussed in section 1.3, except for the *ambiX* suite. Additionally, recordings made using classic Soundfield microphones also use FuMa encoding. To ensure interoperability with these and other sources, processors, and plugin suites, ATK for Reaper uses FuMa FOA signals throughout.

Cartesian coordinates are avoided throughout the plugin suite, as ATK for Reaper strives to provide a consistent and intuitive interface towards parameters regardless of whether the user is accustomed with acoustics or navigational coordinate systems. Whenever possible the plugins offer graphical user interfaces to avoid ambiguities.

Due to current limitations in the Reaper JSFX API, there are some considerations to be made with respect to description of azimuths. The desired behaviour is that azimuths increase anti-clockwise. At the same time it is also preferable that for sources coming from the front half circle, moving the azimuth slider to the right results in the direction of the sound also moving clockwise to the right, similar to what happens when moving a regular stereo pan pot. The Harpex and Blue Ripple plugins both functions this way. In order to achieve this, horizontal azimuth sliders will need to be implemented with increasing values to the left. This is however currently not supported in Reaper JSFX, and for the time being ATK for Reaper uses azimuth values described according to a navigational coordinate system. If Reaper in the future is updated to support sliders with increasing values to the left, this design decision is likely to be reconsidered, and changed to use acoustic conventions.

3.3 Graphical user interfaces in spatial transforms

When a mono source is encoded and the resulting B-format signal is exposed to subsequent matrix-based spatial trans-

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[12] http://www.juce.com/about-juce
[13] https://github.com/olilarkin/wdl-ol
[14] http://www.reaper.fm/sdk/js/js.php

forms as discussed in section 4.2, the resulting B-format signal W, X, Y, Z relates to the original mono signal $s(t)$ as:

$$\begin{aligned}
W(t) &= k_w s(t) \\
X(t) &= k_x s(t) \\
Y(t) &= k_y s(t) \\
Z(t) &= k_z s(t)
\end{aligned} \tag{1}$$

The four coefficients $k_{w,x,y,z}$ indicates four degrees of freedom, and relates to gain g, azimuth ϕ, elevation θ and the degree of directness γ (versus omnipresence, as discussed in section 4.2.2) as:

$$\begin{aligned}
k_w &= g\sqrt{\frac{1 + sin(\gamma)}{2}} \\
k_x &= g\sqrt{1 - sin(\gamma)}\cos\phi\cos\theta \\
k_y &= g\sqrt{1 - sin(\gamma)}\cos\phi\sin\theta \\
k_z &= g\sqrt{1 - sin(\gamma)}\sin\phi
\end{aligned} \tag{2}$$

It follows that if the coefficients $k_{w,x,y,z}$ of the transformed signal are known, gain, azimuth, elevation and directness can be calculated as:

$$\begin{aligned}
\phi &= \operatorname{atan2}(k_y, k_x) \\
\theta &= \operatorname{atan2}(k_z, \sqrt{k_x{}^2 + k_y{}^2}) \\
g &= k_w\sqrt{\frac{2}{1 + \sin\gamma}} \\
\gamma &= \arcsin\left(\frac{2k_w{}^2 - (k_x{}^2 + k_y{}^2 + k_z{}^2)}{2k_w{}^2 + (k_x{}^2 + k_y{}^2 + k_z{}^2)}\right)
\end{aligned} \tag{3}$$

The graphical user interface (GUI) for several of the ATK for Reaper plugins use equation 3 directly to illustrate the effects of the applied spatial transform. A group of hypothetical sources (as planewaves) are placed at equidistant angles around the horizontal plane. These are subsequently processed via the specified transform, and the GUI then displays the transformed prototype source image state.

The soundfield hemisphere is viewed from above. *Gain* level for the transformed sources is indicated by colour hue, with $0\,\mathrm{dB}$ being orange. Signals with increased gain become red, and signals with reduced gain venture towards green and blue. This can be seen in figure 6. The representation of *azimuth* is straight-forward. *Elevation* is indicated by adjustments to lightness and saturation, as well as slight alterations of radius. Additionally, distance from the centre of the hemisphere is reduced with increasing elevation, as seen in figure 2.

With decreasing *directness* (directional definition) the circle grows in size, moves towards the origin, and becomes increasingly transparent. In this way the visualisation helps communicate that an omnipresent source does *not* sound as a 'focused source' located in the centre.[15] This is a common misrepresentation found in 'hockey puck' surround panners. Rather, the sound will appear to be arriving

[15] Interior array 'focused sources' are only possible with very high order HOA or WFS.

from *all* directions. The representation of less directional sources can be seen in figures 3-5.

Depending on the plugin, one or more bright blue knobs in the GUI can be used to control azimuth and degree of transformation.

4. ATK REAPER: THE SUITE OF PLUGINS

ATK for Reaper is implemented as a set of Reaper JSFX source files, one for each plugin. A shared library file containing mathematical constants and conversions, matrix operators and graphics calls used by several of the plugins, ensures a DRY (Don't Repeat Yourself) programming approach. Plugins are sorted into subfolders by categories *Encode*, *Transform* and *Decode*, making it easy for the user to understand the scope of each of the plugins–and understand their roles within the ATK production flow paradigm. The following sections describe these.

4.1 Encoders

Most users approaching Ambisonics are usually presented with two avenues to author an Ambisonic soundfield: capture a natural soundfield directly with a Soundfield microphone, or author a planewave[16] from a monophonic signal. The ATK provides a much wider palette of authoring opportunities.

4.1.1 Basic Encoders

The *Omni* plugin encodes a mono signal as an omnidirectional soundfield. This can be regarded in two ways: a soundfield with no directional information,[17] or a soundfield with an infinite number of planewaves arriving in all directions at once. In a well aligned damped studio environment, this usually sounds 'in the head'. In concert hall listening it usually appears as omnipresent.

A number of planewave encoders are included in the ATK. The *PlaneWave* plugin provides classic encoding of a mono source as a planewave, where the arrival direction (azimuth and elevation) can be set. The *Stereo* plugin encodes stereo left and right channels as two planewaves coming from left and right; the angular spread between the two waves is parameterised. Additional trans-coders are provided for a number of fixed-channel distribution formats: *Quad*, *5.0* and *7.0* for the standard ITU layouts, and *Pantophonic* for 2D and *Periphonic* for 3D arrays.

Perhaps the most *idiomatic* Ambisonic encoder is the ATK's *AtoB*, which provides a powerful method for constructing full, complex soundfields via A-format encoding.[18] The user supplies four separate but related signals to be distributed equally throughout the three dimensions of the soundfield. These signals may either be synthesised or captured by microphones. In FOA, A-format can be visualised as a tetrahedral sampling of the soundfield.[19] A

[16] Planewave encoding is the classic Ambisonic panning technique.
[17] Following the omnidirectional encoder with a *transformer* that adds *directness* is expected for most uses.
[18] The A-format encoder is a simple matrix and does not apply the frequency dependent coincidence compensation filters found in the Soundfield microphone [7].
[19] Up to 3rd order HOA various Platonic solids may be used to sample the soundfield.

variety of tetrahedral orientations are made available.

4.1.2 Advanced Encoders

A few encoders remain to be ported from ATK for Super-Collider: The *PseudoInverse* encoding technique provides a great deal of flexibility, and can be used with both microphone arrays and synthetic signals. In the absence of a Soundfield microphone, this encoding technique gives the opportunity to deploy real-world microphone arrays (omni, cardioid, etc.) to capture natural soundfields. *ZoomH2* adapts *PseudoInverse* for this popular portable recorder.

The final class of encoders remaining to be ported require convolution. These include *Super*, the classic 'super stereo' method for encoding stereophonic signals. The *UHJ* encoder offers access to numerous published recordings for periphonic (2D) audition. *Spread* encodes a monophonic signal by smoothly rotating the signal across the soundfield by frequency where *Diffuse* randomises the phase of the incoming monophonic signal to create a diffuse field. Along with *AtoB*, these last two are regarded as the primary 'full' soundfield encoding tools to be found in the ATK.

We look forward to adding all these encoders soon in the JSFX distribution.

4.2 Imaging transforms

For the artist, the real power of the ATK is found in the imaging transforms. These are spatial domain filters which reorient, reshape or otherwise spatially filter an input soundfield. As discussed in section 1.3, there are other plugin libraries available for spatial transforms. The ATK provides a much wider and comprehensive toolset.

As described earlier and illustrated in figure 1, it is expected that Authoring (*encoders*), will be followed by Imaging (*transformers*). One can regard this as augmenting the panning law of the initial encoding. For example, an omnidirectional or diffuse soundfield may be 'pushed' into a planewave arriving from a single direction. Or, conversely, the directivity of a soundfield composed of planewaves arriving from many directions may be collapsed to a directionless (omnidirectional) field.

4.2.1 RotateTiltTumble

The *RotateTiltTumble* plugin provides multi-axes FOA rotations, with *Rotation*, *Tilt* and *Tumble* [20] applied in the sequence indicated by the plugin name. If the user wishes to change the order of rotations, it is possible to daisy-chain two or more instances of the plugin. Rotation does not affect directivity of the signal. The screenshot in figure 2 demonstrates how variations in saturation, lightness and radius of the displayed transformed sources serve to illustrate their vertical position. The separate effect of the *Tilt* and *Tumble* transformations is indicated by the two blue planes in the interface.

4.2.2 Direct and DirectO

The *DirectO* plugin (figure 3) adjusts the directivity of an FOA soundfield across the origin. It is a spatial low-pass
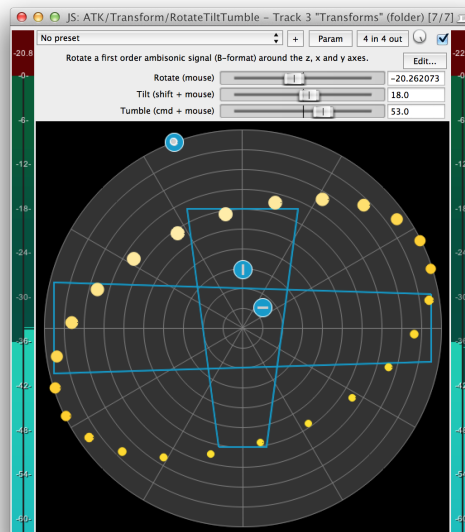
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[20] Yaw, Roll, and Pitch

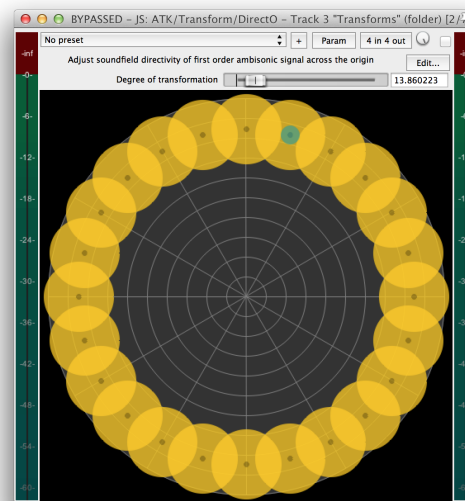**Figure 2**: Rotate, Tilt and Tumble transform.

Figure 3: DirectO transform plugin interface.

filter; with with an increasing degree of transformation, the soundfield becomes less directional, and with a transform of $90°$ the soundfield becomes omnipresent.

Similarly, the *Direct* plugin (figure 4) adjusts the soundfield directivity across across a plane specified by the user.

4.2.3 FocusPressPushZoom and Dominate

The *FocusPressPushZoom* plugin provides a unified interface to four different spatial transforms: *Focus*, *Press*, *Push* and *Zoom*. These primarily differ in the directional response of the target planewave and the scaling of gain across the soundfield. The amount of spatial transform is expressed as a soundfield *distortion angle* [21] in the range from $0°$ to $90°$. At $0°$ no distortion is applied. At $90°$ all four transforms collapse the soundfield to a planewave arriving from the direction of interest specified by *azimuth*

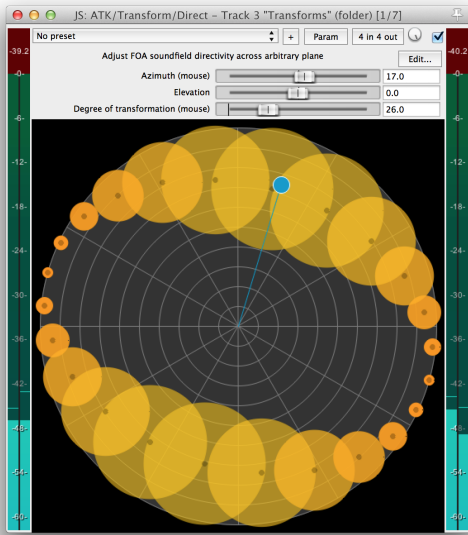[21] Auditioned as a warping of the soundfield.

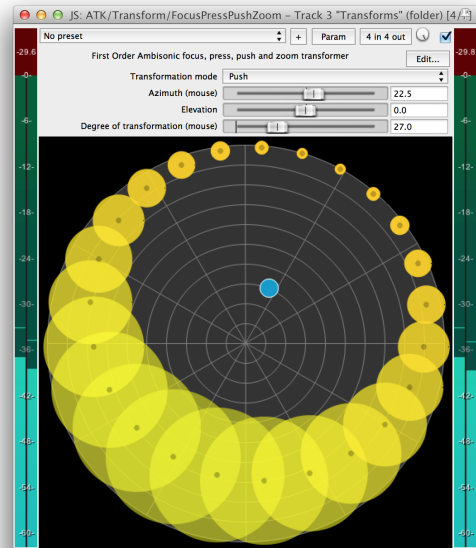**Figure 4**: Direct transform plugin interface.

Figure 5: Push transform plugin interface.

and *elevation*. These transforms fall into two groups.

Focus and *Zoom* are dominance related transforms and can be described as 'emphasising' elements in the direction of interest. The directional response of the target planewave (*distortion angle* 90°) is a cardioid. This means that gain opposite the direction of interest is reduced. *Focus* maintains the gain forward at 0 dB. *Zoom*, however, increases gain in the direction of interest and maintains gain perpendicular to this direction.

Press and *Push* [22] act differently. Rather than emphasising elements in a target direction, all are 'pressed' or 'pushed' towards the direction of interest. Where the target for *Focus* and *Zoom* is a cardioid, the target for this second group is an omnidirectional response. As a result, the gains of all elements are maintained to a similar level and gathered toward the direction of interest. With all four transforms available in the same plugin it becomes easy to explore spatial expressive quality as part of the creative process. GUIs for two can be seen in figures 5 and 6.

The *Dominance* plugin increases the gain of elements in the direction of interest while decreasing gain opposite. With the JSFX plugin *directional gain* can be boosted by up to 24 dB. While the *Focus*, *Press*, *Push*, and *Zoom* transforms do not generally have a significant impact on the overall sound level of the soundfield, *Dominance* can result in major alterations to sound levels. For this reason, and the difference in the way the amount of soundfield transformation is specified (gain in dB vs. distortion angle in degrees), the *Dominance* transform is implemented as a separate plugin.

4.2.4 Mirroring

The *Mirror* plugin mirrors the soundfield across an arbitrary plane, while the *MirrorO* plugin mirrors the soundfield through the origin.

4.3 Near-field effects

Proximity facilitates the introduction of the proximity effect to encoded signals. The proximity effect can be an important contributor to perceptions of nearness. At extremes, the proximity effect introduces a strong bass boost, requiring careful handling. Distance is described in meters, and to prevent bass boost from getting out of hand, the lower limit is set to 10 cm (0.1 m).

Nearfield compensation (NFC) is implemented via the *NearFieldCompensation* plugin, and facilitates the reduction or removal of the proximity effect from encoded signals. NFC is usually used in conjunction with decoders to compensate for the distance of loudspeakers on playback in loudspeaker rigs with a smaller diameter. For the artist, NFC can also be used to reduce the proximity effect found in nearfield natural recordings.

Proximity and *NearFieldCompensation* respectively apply an integrator or high pass filter to the first order components of the encoded signal. [23] The *Proximity* filter undoes *NearFieldCompensation* given the same distance argument.

4.4 Audio Effects and Ambisonics

Spatial information is encoded in balance and phase relations between the four channels of a B-format signal. Processing one or more of the B-format channels using an audio effect which modifies gain and phase relations will most likely disrupt or otherwise distort the encoded spatial information. In most cases this is un-desired. The ATK provides a remedy through the use of a pair of plugins, *BtoA* and *AtoB*.

Native B-format signals are processed using familiar audio effects by first converting to A-format. Figure 7 illustrates the appropriate network. The *BtoA* plugin decodes

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[22] Two novel spatial transforms, at present found only in the ATK.

[23] A 1st order high pass filter is the reciprocal of a 1st order integrator.
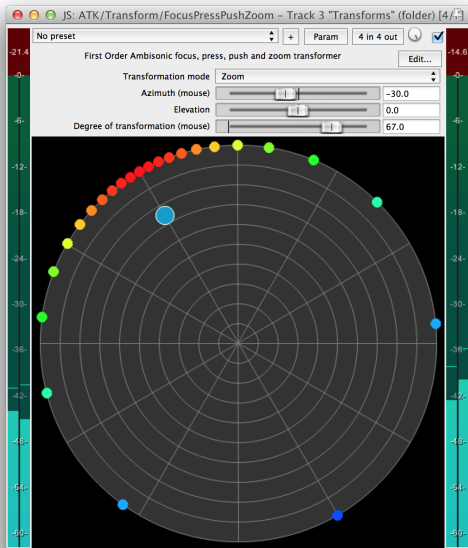
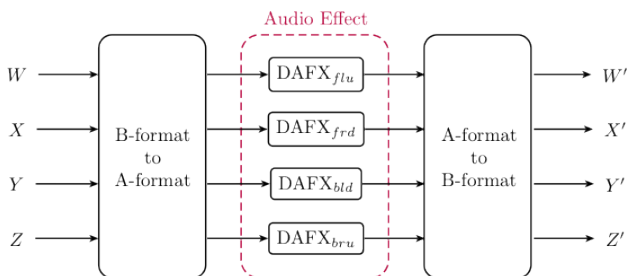**Figure 6**: Zoom transform interface.

Figure 7: Process effects in A-format.

to A-format, while *AtoB* re-encodes into B-format. Spatial information is preserved (or modified) in a similar way as effects processing with two-channel stereo.

The *Omni* mono signal encoder in conjunction with the network illustrated in figure 7 can also be used to synthesise interesting soundfields. Each of the effects units should be tuned with slightly different parameters to ensure an active soundfield for this application.

4.5 Decoders

Perhaps one of the most celebrated aspects of the Ambisonic sound technique has been its design as a hierarchal reproduction system, able to target a number of varying loudspeaker arrays. The ATK provides a wide palette of optimised decoders.

The *Mono* virtual microphone decoder returns a single channel, and can be used to 'listen in' to the soundfield at the specified azimuth and elevation. [24] The *Stereo* decoder returns a pair of virtual microphones. Virtual microphone polar patterns may be specified for both decoders.

Surround array decoders are included in a number of forms. *Quad* is an optimised quadraphonic decoder with variable loudspeaker angle, *5.0* uses Wiggins optimised ITU 5.0

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[24] This can also be thought of as extracting a monophonic feed from a soundfield, e.g., a natural soundfield recording, in a direction of interest.

decoders [20], *Pantophonic* is a regular 2D polygon decoder and *Periphonic* is a regular cylindrical decoder for 3D dual ring. *Diametric* is Gerzon's classic decoder suitable for varied periphonic and pantophonic loudspeaker arrays [21]. While regular decoders are suitable for many users, diametric decoding enables the greatest flexibility, and allows the design of substantially varying semi-regular arrays suitable for a wide variety of playback situations. Meeting all the criteria outlined by Gerzon to qualify as Ambisonic, this decoder is a good choice for full 3D critical studio listening.

The above surround decoders supports several modes. A two-band psycho-acoustically optimised decoder is returned by *dual*, the optimum choice for small scale studio or domestic settings [22]. Suitable for mid-scale arrays is the *single* (aka 'energy') decoder. 'Controlled opposites' decoding (aka 'in phase') is returned by *controlled*, and may be preferred for large scale arrays. The *velocity* setting returns 'strict soundfield' (aka 'basic') decoding. [25]

The *UHJ* decoder and binaural decoders, using measured and synthetic HRTFs, require kernel convolution. These remain to be ported. While the virtual microphone stereophonic decoder is very easy and convenient, for production work the authors advise using the *UHJ* decoder once the port has been accomplished.

5. DISCUSSION

While there is a well-established workflow for stereo on DAWs, options have been more limited when working with Ambisonics. From a compositional viewpoint, Ambisonic field recordings sometimes can leave the sonic artist with an impression of 'take it or leave it', as the sound image can be both spectrally and spatially quite 'full'. ATK for Reaper brings both a comprehensive composition workflow model, the *sound-field sound-image* paradigm, and a set of versatile tools for working with Ambisonics to the DAW environment. The imaging tools available allow soundfield imaging to become part of a compositional language: naturally recorded 'too full' soundfields can be reduced and controlled to fit a creative context, 'too small' synthetic generated material can be expanded.

In addition to Reaper's flexible channel routing and signal flow, the *FX Chains* preset plugin chain feature accommodates the ATK's design paradigm very well, and allows the user to assemble more complex Ambisonic signal processing by cascading a number of ATK plugins. These can then be saved as *FX Chains* to be used again. A quadraphonic decoder with nearfield compensation is a simple example. The user would create an *FX Chain* consisting of: *NearFieldCompensation→Quad*. A more complex example could be a 'custom DAFX' mono encoder. Expanding on the network illustrated in figure 7: *Omni→BtoA→DAFX→AtoB→Push→Proximity*. As a saved *FX Chain*, this DAFX processor-panner then becomes available throughout a user workflow, and mirrors the method found in ATK for SuperCollider. Similarly, favourite DAFX processes used in stereo workflows

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[25] Usually not preferred for FOA in most cases.

can be adapted to B-format as discussed in section 4.4 and saved as *FX Chains* for re-use.

ATK for SuperCollider3 is distributed as open-source using the GNU GPL General Public License Version 3. This license is incompatible for use with proprietary software such as Reaper, and for this reason ATK for Reaper is distributed using the GNU LGPL Lesser Public License Version 3. ATK for Reaper will be available for download from the Ambisonic Toolkit web site in the near future. [26]

Acknowledgments

6. REFERENCES

[1] ITU, "Recommendation BS. 775: Multi-channel stereophonic sound system with or without accompanying picture," International Telecommunications Union, Tech. Rep., 1993.

[2] Dolby Laboratories, Inc, "Dolby Atmos. Next-generation audio for cinema. White paper Issue 3," Dolby Laboratories, Inc, Tech. Rep., 2014. [Online]. Available: http://www.dolby.com/uploadedFiles/Assets/US/Doc/Professional/Dolby-Atmos-Next-Generation-Audio-for-Cinema.pdf

[3] A. Berkhout, D. Vries, and P. Vogel, "Acoustic Control by Wave Field Synthesis," *J. Acoust. Soc. Am.*, vol. 93, pp. 2764–2778, 1993.

[4] N. Peters, T. Lossius, and J. Schacher, "The Spatial Sound Description Interchange Format: Principles, specification, and examples," *Computer Music Journal*, vol. 37, no. 1, pp. 11–22, Spring 2013.

[5] M. A. Gerzon, "Ambisonics in multichannel broadcasting and video," *J. Audio Eng. Soc.*, vol. 3, no. 11, pp. 859–871, Nov. 1985.

[6] J. Daniel, "Représentation de champs acoustiques, application à la transmission et à la reproduction de scènes sonores complexes dans un contexte multimédia," Ph.D. dissertation, Université Paris 6, 2001.

[7] K. Farrar, "Soundfield microphone: design and development of microphone and control unit," *Wireless World*, pp. 48–50 (Oct.), 99–103 (Nov.), 1979.

[8] IOsONO GmbH, "Spatial audio workstation 2 operation manual," IOsONO GmbH, Tech. Rep., 2012.

[9] Dolby Laboratories, Inc, "Authoring for Dolby Atmos cinema sound manual," Dolby Laboratories, Tech. Rep., 2013. [Online]. Available: http://www.dolby.com/uploadedFiles/Assets/US/Doc/Professional/Authoring_for_Dolby_Atmos_Cinema_Sound_Manual(1).pdf

[10] G. Francis, "Up and running: A REAPER user guide," Cockos, http://reaper.fm/userguide.php, Tech. Rep., March 2014.

[11] M. Kronlachner, "Ambisonics plug-in suite for production and performance usage," in *Proceedings of the Linux Audio Conference, Graz*, 2013.

[12] M. Kronlachner and F. Zotter, "Spatial transformations for the enhancement of ambisonic recordings," in *Proceedings of the 2nd International Conference on Spatial Audio, Erlangen*, 2014.

[13] C. Nachbar, F. Zotter, E. Deleflie, and A. Sontacchi, "Ambix - a suggested ambisonics format," in *Proceedings of the Ambisomics Symposium, Lexington*, 2011.

[14] M. Kronlachner, "Spatial transformations for the enhancement of ambisonic recordings," Master's thesis, IEM Institute of Electronic Music and Acoustics, Graz, 2014.

[15] IRCAM and Flux, "Spat multiformat room acoustics simulation and localization processor. manual," IRCAM and Flux, Tech. Rep., 2010.

[16] S. Berge and N. Barrett, "High angular resolution planewave expansion," in *Proceedings of the 2nd International Symposium on Ambisonics and Spherical Acoustics, Paris*, 2010.

[17] J. Anderson, "Introducing...the Ambisonic Toolkit," in *Proceedings of the Ambisonics Symposium, Graz*, 2009.

[18] Apple Inc., "Audio Unit programming guide," Apple Inc., Tech. Rep., 2014. [Online]. Available: https://developer.apple.com/library/mac/documentation/MusicAudio/Conceptual/AudioUnitProgrammingGuide/Introduction/Introduction.html

[19] J. Smith III, *Audio Signal Processing in Faust*, Center for Computer Research in Music and Acoustics (CCRMA), Department of Music, Stanford University, Section 4, 2012. [Online]. Available: http://ccrma.stanford.edu/~jos/aspf/aspf.pdf

[20] B. Wiggins, I. Paterson-Stephens, V. Lowndes, and S. Berry, "The design and optimisation of surround sound decoders using heuristic methods," in *Proceedings of UKSIM 2003: Conference on Computer Simulation*, Cambridge, England, 2003.

[21] M. A. Gerzon, "Practical periphony: The reproduction of full-sphere sound," in *Proceedings of the 65th Audio Engineering Engineering Society Convention*, London, 1980.

[22] A. Heller, R. Lee, and E. Benjamin, "Is my decoder ambisonic?" in *Proceedings of the 125th Audio Engineering Society Convention*, San Francisco, 2008.

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[26] http://www.ambisonictoolkit.net